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# UTILITY PATENT APPLICATION TRANSMITTAL

(Only for new nonprovisional applications under 37 C.F.R. § 1.53(b))

Attorney Docket No. 15-0195

First Inventor or Application Identifier RUSSELL H. LAMBERT

Title SEE ATTACHED ADDENDUM I

Express Mail Label No. EJ789028426US

## APPLICATION ELEMENTS

See MPEP chapter 600 concerning utility patent application contents.

1. ☒ \* Fee Transmittal Form (e.g., PTO/SB/17)  
(Submit an original and a duplicate for fee processing)
2. ☒ Specification [Total Pages 17]  
(preferred arrangement set forth below)
  - Descriptive title of the invention
  - Cross References to Related Applications
  - Statement Regarding Fed sponsored R & D
  - Reference to Microfiche Appendix
  - Background of the invention
  - Brief Summary of the invention
  - Brief Description of the Drawings (if filed)
  - Detailed Description
  - Claim(s)
  - Abstract of the Disclosure
3. ☒ Drawing(s) (35 U.S.C. 113) [Total Sheets 5]
4. Oath or Declaration [Total Pages 3]
  - a. ☒ Newly executed (original or copy)
  - b. ☐ Copy from a prior application (37 C.F.R. § 1.63(d))  
(for continuation/divisional with Box 16 completed)
    - i. ☐ **DELETION OF INVENTOR(S)**  
Signed statement attached deleting  
inventor(s) named in the prior application,  
see 37 C.F.R. §§ 1.63(d)(2) and 1.33(b).

**NOTE FOR ITEMS 1 & 13 IN ORDER TO BE ENTITLED TO PAY SMALL ENTITY FEES, A SMALL ENTITY STATEMENT IS REQUIRED (37 C.F.R. § 1.27), EXCEPT IF ONE FILED IN A PRIOR APPLICATION IS RELIED UPON (37 C.F.R. § 1.28).**

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5. ☐ Microfiche Computer Program (Appendix)
6. Nucleotide and/or Amino Acid Sequence Submission  
(if applicable, all necessary)
  - a. ☐ Computer Readable Copy
  - b. ☐ Paper Copy (identical to computer copy)
  - c. ☐ Statement verifying identity of above copies

## ACCOMPANYING APPLICATION PARTS

7. ☒ Assignment Papers (cover sheet & document(s))
8. ☒ 37 C.F.R. § 3.73(b) Statement ☒ Power of  
(when there is an assignee) Attorney
9. ☐ English Translation Document (if applicable)
10. ☒ Information Disclosure ☒ Copies of IDS  
Statement (IDS)/PTO-1449 Citations
11. ☐ Preliminary Amendment
12. ☒ Return Receipt Postcard (MPEP 503)  
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13. ☐ \* Small Entity ☐ Statement filed in prior application  
Statement(s) Status still proper and desired  
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14. ☐ Certified Copy of Priority Document(s)  
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16. If a CONTINUING APPLICATION, check appropriate box, and supply the requisite information below and in a preliminary amendment:

☐ Continuation ☐ Divisional ☐ Continuation-in-part (CIP)

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TRW Docket No. 15-0195

UTILITY PATENT APPLICATION TRANSMITTAL

(Addendum 1)

TITLE: MICROPHONE ARRAY PROCESSING SYSTEM  
FOR NOISY MULTIPATH ENVIRONMENTS

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## MICROPHONE ARRAY PROCESSING SYSTEM FOR NOISY MULTIPATH ENVIRONMENTS

### BACKGROUND OF THE INVENTION

5

This invention relates generally to techniques for reliable conversion of speech data from acoustic signals to electrical signals in an acoustically noisy and reverberant environment. There is a growing demand for "hands-free" cellular telephone communication from automobiles, using automatic speech recognition (ASR) for dialing and other functions. However, background noise from both inside and outside an automobile renders in-vehicle communication both difficult and stressful. Reverberation within the automobile combines with high noise levels to greatly degrade the speech signal received by a microphone in the automobile. The microphone receives not only the original speech signal but also distorted and delayed duplicates of the speech signal, generated by multiple echoes from walls, windows and objects in the automobile interior. These duplicate signals in general arrive at the microphone over different paths. Hence the term "multipath" is often applied to the environment. The quality of the speech signal is extremely degraded in such an environment, and the accuracy of any associated ASR systems is also degraded, perhaps to the point where they no longer operate. For example, recognition accuracy of ASR systems as high as 96% in a quiet environment could drop to well below 50% in a moving automobile.

Another related technology affected by a noise and reverberation is speech compression, which digitally encodes speech signals to achieve reductions in communication bandwidth and for other reasons. In the presence of noise, speech compression becomes increasingly difficult and unreliable.

In the prior art, sensor arrays have been used or suggested for processing narrowband signals, usually with a fixed uniformly spaced microphone

array, with each microphone having a single weighting coefficient. There are also wideband array signal processing systems for speech applications. They use a beam-steering technique to position "nulls" in the direction of noise or jamming sources. This only works, of course, if the noise is emanating from one or a small  
5 number of point sources. In a reverberant or multipath environment, the noise appears to emanate from many different directions, so noise nulling by conventional beam steering is not a practical solution.

There are also a number of prior art systems that effect active noise cancellation in the acoustic field. Basically, this technique cancels acoustic noise  
10 signals by generating an opposite signal, sometimes referred to as "anti-noise," through one or more transducers near the noise source, to cancel the unwanted noise signal. This technique often creates noise at some other location in the vicinity of the speaker, and is not a practical solution for canceling multiple unknown noise sources, especially in the presence of multipath effects.

Accordingly, there is still a significant need for reduction of the effects  
15 of noise in a reverberant environment, such as the interior of a moving automobile. As discussed in the following summary, the present invention addresses this need.

## **SUMMARY OF THE INVENTION**

The present invention resides in a system and related method for noise  
20 reduction in a reverberant environment, such as an automobile. Briefly, and in general terms, the system of the invention comprises a plurality of microphones positioned to detect speech from a single speech source and noise from multiple  
25 sources, and to generate corresponding microphone output signals, one of the microphones being designated a reference microphone and the others being designated data microphones. The system further comprises a plurality of bandpass filters, one for each microphone, for eliminating from the microphone output signals a known spectral band containing noise; a plurality of adaptive filters, one for each of  
30 the data microphones, for aligning each data microphone output signal with the

output signal from the reference microphone; and a signal summation circuit, for combining the filtered output signals from the microphones. Signal components resulting from the speech source combine coherently and signal components resulting from multiple noise sources combine incoherently, to produce an increased  
5 signal-to-noise ratio. The system may also comprise speech conditioning circuitry coupled to the signal summation circuit, to reduce reverberation effects in the output signal.

More specifically, each of the adaptive filters includes means for filtering data microphone output signals by convolution with a vector of weight  
10 values; means for comparing the filtered data microphone output signals from one of the data microphones with reference microphone output signals and deriving therefrom an error signal; and means for adjusting the weight values convolved with the data microphone output signals to minimize the error signal. In the preferred embodiment of the invention, each of the adaptive filters further includes fast Fourier  
15 transform means, to transform successive blocks of data microphone output signals to a frequency domain representation to facilitate real-time adaptive filtering.

The invention may also be defined in terms of a method for improving detection of speech signals in noisy environments. Briefly, the method comprises the steps of positioning a plurality of microphones to detect speech from a single speech  
20 source and noise from multiple sources, one of the microphones being designated a reference microphone and the others being designated data microphones; generating microphone output signals in the microphones; filtering the microphone output signals in a plurality of bandpass filters, one for each microphone, to eliminate from the microphone output signals a known spectral band containing noise;  
25 adaptively filtering the microphone output signals in a plurality of adaptive filters, one for each of the data microphones, and thereby aligning each data microphone output signal with the output signal from the reference microphone; and combining the adaptively filtered output signals from the microphones in a signal summation circuit. The incoming speech from one or multiple microphones is monitored to determine  
30 when speech is present. The adaptive filters are only allowed to adapt while speech

is present. Signal components resulting from the speech source combine coherently in the signal summation circuit and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio. The method may further comprise the step of conditioning the combined signals in speech conditioning  
5 circuitry coupled to the signal summation circuit, to reduce reverberation effects in the output signal.

More specifically, the step of adaptively filtering includes filtering data microphone output signals by convolution with a vector of weight values; comparing the filtered data microphone output signals from one of the data microphones with  
10 reference microphone output signals and deriving therefrom an error signal; adjusting the weight values convolved with the data microphone output signals to minimize the error signal; and repeating the filtering, comparing and adjusting steps to converge on a set of weight values that results in minimization of noise effects.

In the preferred embodiment of the invention, the step of adaptively  
15 filtering further includes obtaining a block of data microphone signals; transforming the block of data to a frequency domain using a fast Fourier transform; filtering the block of data in the frequency domain using a current best estimate of weighting values; comparing the filtered block of data with corresponding data derived from the reference microphone; updating the filter weight values to minimize any difference  
20 detected in the comparing step; transforming the filter weight values back to the time domain using an inverse fast Fourier transform; zeroing out portions of the filter weight values that give rise to unwanted circular convolution; and converting the filter values back to the frequency domain.

It will be appreciated from the foregoing summary that the present  
25 invention represents a significant advance in speech communication techniques, and more specifically in techniques for enhancing the quality of speech signals produced in a noisy environment. The invention improves signal-to-noise performance and reduces the reverberation effects, providing speech signals that are more intelligible to users. The invention also improves the accuracy of automatic speech recognition  
30 systems. Other aspects and advantages of the invention will become apparent from

the following more detailed description, taken in conjunction with the accompanying drawings.

## BRIEF DESCRIPTION OF THE DRAWINGS

5

FIGURE 1 is a block diagram depicting an important aspect of the invention, wherein signal amplitude is increased by coherent addition of filtered signals from multiple microphones;

FIG. 2 is another block diagram showing a microphone array in accordance with the invention, and including bandpass filters, speech detection circuitry, adaptive filters, a signal summation circuit, and speech conditioning circuitry;

FIGS. 3A and 3B together depict another block diagram of the invention, including more detail of adaptive filters coupled to receive microphone outputs;

FIG. 4 is a block diagram showing detail of a single adaptive filter used in the invention;

FIG. 5 is another block diagram of the invention, showing how noise signal components are effectively reduced in accordance with the invention;

FIG. 6 is a graph showing a composite output signal from a single microphone detecting a single speaker in a noisy automobile environment; and

FIG. 7 is a graph showing a composite output signal obtained from an array of seven microphones in accordance with the invention, while processing speech from a single speaker in conditions similar to those encountered in the generation of the graph of FIG. 6.

## DESCRIPTION OF THE PREFERRED EMBODIMENTS

As shown in the drawings, the present invention is concerned with a technique for significantly reducing the effects of noise in the detection or recognition

of speech in a noisy and reverberant environment, such as the interior of a moving automobile. The quality of speech transmission from mobile telephones in automobiles has long been known to be poor much of the time. Noise from within and outside the vehicle result in a relatively low signal-to-noise ratio and  
 5 reverberation of sounds within the vehicle further degrades the speech signals. Available technologies for automatic speech recognition (ASR) and speech compression are at best degraded, and may not operate at all in the environment of the automobile.

In accordance with the present invention, use of an array of  
 10 microphones and its associated processing system results in a significant improvement in signal-to-noise ratio, which enhances the quality of the transmitted voice signals, and facilitates the successful implementation of such technologies as ASR and speech compression.

The present invention operates on the assumption that noise emanates  
 15 from many directions. In a moving automobile, noise sources inside and outside the vehicle clearly do emanate from different directions. Moreover, after multiple reflections inside the vehicle, even noise from a point source reaches a microphone from multiple directions. A source of speech, however, is assumed to be a point source that does not move, at least not rapidly. Since the noise comes from many  
 20 directions it is largely independent, or uncorrelated, at each microphone. The system of the invention sums signals from N microphones and, in so doing, achieves a power gain of  $N^2$  for the signal of interest, because the amplitudes of the individual signals from the microphones sum coherently, and power is proportional to the square of the amplitude. Because the noise components obtained from the  
 25 microphones are incoherent, summing them together results in an incoherent power gain proportional to N. Therefore, there is a signal-to-noise ratio improvement by a factor of  $N^2/N$ , or N.

FIG. 1 shows an array of three microphones, indicated at 10.1, 10.2 and 10.3, respectively. Microphone 10.1 is designated the reference microphone and  
 30 the other two microphones are designated data microphones. Each microphone



receives an acoustic signal  $S$  from a speech source 12. For purposes of explanation, in this illustration noise is considered to be absent. The acoustic transfer functions for the three microphones are  $h_1$ ,  $h_2$  and  $h_3$ , respectively. Thus, the electrical output signals from the microphones are  $S \cdot h_1$ ,  $S \cdot h_2$  and  $S \cdot h_3$ , respectively. The signals from the data microphones 10.2 and 10.3 are processed as shown in blocks 14 and 16, respectively, to allow them to be combined with each other and with the reference microphone signal. In block 14, the acoustic path transfer function  $h_2$  is inverted and the reference acoustic path transfer function  $h_1$  is applied, to yield the signal  $S \cdot h_1$ . Similarly, in block 16, the function  $h_3$  is inverted and the function  $h_1$  is applied, to yield the signal  $S \cdot h_1$ . The three microphone signals are then applied to a summation circuit 18, which yields at output of  $3 \cdot S \cdot h_1$ . This signal is then processed by speech conditioning circuitry 20, which effectively inverts the transfer function  $h_1$  and yields the resulting signal amplitude  $3S$ . An array of  $N$  microphones would yield an effective signal amplitude gain of  $N$  (a power gain of  $N^2$ ).

The incoming speech to one or multiple microphones 10 is monitored in speech detection circuitry 21 to determine when speech is present. The functions performed in blocks 14 and 16 are performed only when speech is detected by the circuitry 21.

The signal gain obtained from the array of microphones is not dependent in any way on the geometry of the array. One requirement for positioning the microphones is that they be close enough to the speech source to provide a strong signal. A second requirement is that the microphones be spatially separated. This spatial separation is needed so that independent noises are sampled. Similarly, noise reduction in accordance with the invention is not dependent on the geometry of the microphone array.

The purpose of the speech conditioning circuitry 20 is to modify the spectrum of the cumulative signal obtained from the summation circuit 18 to resemble the spectrum of "clean" speech obtained in ideal conditions. The amplified signal obtained from the summation circuit 18 is still a reverberated one. Some improvement is obtained by equalizing the magnitude spectrum of the output signal

to match a typical representative clean speech spectrum. A simple implementation of the speech conditioning circuitry 20, therefore, includes an equalizer that selectively amplifies spectral bands of the output signal to render the spectrum consistent with the clear speech spectrum. A more advanced form of speech conditioning circuitry is

5 a blind equalization process specially tailored for speech. (See, for example, Lambert, R.H. and Nikias, C.L., "Blind Deconvolution of Multipath Mixtures," Chapter from *Unsupervised Adaptive Filtering*, Vol. 1, edited by Simon Haykin, John Wiley & Sons, 1999.) This speech conditioning process is particularly important when an ASR system is "trained" using clean speech samples. Optimum results are obtained

10 by training the ASR system using the output of the present invention under typical noisy environmental conditions.

FIG. 2 depicts the invention in principle, showing the speech source 12, a reference microphone 10.R, and N data microphones indicated at 10.1 through 10.N. The output from the reference microphone 10.R is coupled to a bandpass filter

15 22.R and the outputs from the data microphones 10.1 through 10.N are coupled to similar bandpass filters 22.1 through 22.N, respectively. A great deal of environmental noise lies in the low frequency region of approximately 0-300 Hz. Therefore, it is advantageous to remove energy in this region to provide an improvement in signal-to-noise ratio.

20 The outputs of the bandpass filters 22.1 through 22.N are connected to adaptive filters 24.1 through 24.N, respectively, indicated in the figure as  $W_1$  through  $W_N$ , respectively. These filters are functionally equivalent to the filters 14 and 16 in FIG. 1. The outputs of the filters 24, indicated as values  $X_1$  through  $X_N$ , are input to the summation circuit 18, the output of which is processed by speech conditioning

25 circuitry 20, as discussed with reference to FIG. 1. As indicated by the arrow 26, output signals from the reference bandpass filter 22.R are used to update the filters  $W_1$  through  $W_N$  periodically, as will be discussed with reference to FIGS. 3 and 4. Speech detection circuitry 21 enables the filters 24 only when speech is detected.

FIGS. 3A and 3B show the configuration of FIG. 2 in more detail, but

30 without the bandpass filters 22 of FIG. 2. FIG. 3A shows the same basic

configuration of microphones 10R and 10.1 through 10.N, each receiving acoustic signals from the speech source 12. FIG. 3B shows the filters  $W_i$  24.1 through  $W_N$  24.N in relation to incoming signals  $y_i$  through  $y_N$  from the data microphones 10.1 through 10.N. Each of the  $W$  filters 24.1 through 24.N has an associated summing circuit 28.1 through 28.N connected to its output. In each summing circuit, the output of the  $W$  filter 24 is subtracted from a signal from the reference microphone 22.R transmitted over line 30 to each of the summing circuits. The result is an error signal that is fed back to the corresponding  $W$  filter 24, which is continually adapted to minimize the error signal.

FIG. 4 shows this filter adaptation process in general terms, wherein the  $i^{\text{th}}$  filter  $W_i$  is shown as processing the output signal from the  $i^{\text{th}}$  data microphone. Adaptive filtering follows conventional techniques for implementing finite impulse response (FIR) filters and can be performed in either the time domain or the frequency domain. In the usual time domain implementation of an adaptive filter,  $W_i$  is a weight vector, representing weighting factors applied to successive outputs of a tapped delay line that forms a transversal filter. In a conventional LMS adaptive filter, the weights of the filter determine its impulse response, and are adaptively updated in the LMS algorithm. Frequency domain implementations have also been proposed, and in general require less computation than the time domain approach. In a frequency domain approach, it is convenient to group the data into blocks and to modify the filter weights only after processing each block.

In the preferred embodiment of the invention, the adaptive filter process is a block frequency domain LMS (least mean squares) adaptive update procedure similar to that described in a paper by E.A. Ferrara, entitled "Fast Implementation of LMS Adaptive Filters," IEEE Trans. On Acoustics, Speech and Signal Processing, Vol. ASSP-28, No. 4, 1980, pp 474-475. The error signal computed in summing circuit 28.i is given by (Reference mic.)  $-y_i * W_i$ . In digital processing of successive blocks of data, one adaptive step of  $W_i$  may be represented by the expression:

$$W_i(k+1) = W_i(k) + \mu(\text{REF}(k) - y_i * W_i(k)) * \text{conj}(Y_i(k)),$$

where  $k$  is the data block number and  $\mu$  is a small adaptive step.

The process described by Ferrara has been modified to provide greater efficiency in a real-time system. The modification entails converting the filters to the time domain, zeroing the portions of the filters that give rise to circular convolution, and then returning the filters to the frequency domain. More specifically, for each data block  $k$ , the following steps are performed:

- Obtain a block of data from the reference microphone and convert the data to the frequency domain.  $\text{REF}(k) = \text{fft}(\text{ref}(k))$ . New data read in is less than one-half of the FFT (fast Fourier transform) size, following a conventional process known as the overlap and save method.
- For each sensor  $i=1$  to  $N$ , perform the following steps:
  - Obtain a block of data  $y_i(k)$  from microphone  $i$  and transform it to the frequency domain.  $Y_i(k) = \text{fft}(y_i(k))$ .
  - Filter the frequency domain block with the current best estimate of  $w_i$  to obtain  $X_i(k) = W_i(k) * Y_i(k)$ .
  - Update the filter using  $W_i(k+1) = W_i(k) + \mu(\text{REF}(k) - X_i(k)) * \text{conj}(Y_i)$ .
  - Convert the frequency domain filter back to the time domain.  $W_i(k+1) = \text{ifft}(W_i(k+1))$ .
  - Zero out portions of  $w_i(k+1)$ .
  - Convert back to the frequency domain.  $W_i(k+1) = \text{fft}(w_i(k+1))$ .

FIG. 5 shows the system of the invention processing speech from the source 12 and noise from multiple sources referred to generally by reference numeral 32. In the summation circuit 18, the speech signal contributions from the data microphones are added coherently, as previously discussed, to produce a speech signal proportional to  $N \cdot S * h_1$ , and this signal can be conveniently convolved with the transfer function  $h_1$  to produce a larger speech signal  $N \cdot S$ . The speech signals, being coherent, combine in amplitude, and since the power of a sinusoidal signal is proportional to the square of its amplitude, the speech signal power from  $N$

sensors will be  $N^2$  times the power from a single sensor. In contrast, the noise components sensed by each microphone come from many different directions, and combine incoherently in the summation circuit 18. The noise components may be represented by the summation:  $n_1 + n_2 + \dots + n_N$ . Because these contributions are incoherent, their powers combine as  $N$  but their root mean square (RMS) amplitudes combine as  $\sqrt{N}$ . The cumulative noise power from the  $N$  sensors is, therefore, increased by a factor  $N$ , and the signal-to-noise ratio (the ratio of signal power to noise power) is increased by a factor  $N^2/N$ , or  $N$ . As in the previously described embodiments of the invention, speech detection circuitry 21 enables the filters 24 only when speech is detected by the circuitry.

Theoretically, if the number of sensors is doubled the single-to-noise ratio should also double, i.e. show an improvement of 3 dB (decibels). In practice, the noise is not perfectly independent at each microphone, so the signal-to-noise ratio improvement obtained from using  $N$  microphones will be somewhat less than  $N$ .

The effect of the adaptive filters in the system of the invention is to "focus" the system on a spherical field surrounding the source of the speech signals. Other sources outside this sphere tend to be eliminated from consideration and noise sources from multiple sources are reduced in effect because they are combined incoherently in the system. In an automobile environment, the system re-adapts in a few seconds when there is a physical change in the environment, such as when passengers enter or leave the vehicle, or luggage items are moved, or when a window is opened or closed.

FIGS. 6 and 7 show the improvement obtained by use of the invention. A composite output signal derived from a single microphone is shown in FIG. 6 and is clearly more noisy than a similar signal derived from seven microphones in accordance with the invention.

It will be appreciated from the foregoing that the present invention represents a significant advance in the field of microphone signal processing in noisy environments. The system of the invention adaptively filters the outputs of multiple

microphones to align their signals with a common reference and allow signal components from a single source to combine coherently, while signal components from multiple noise sources combine incoherently and have a reduced effect. The effect of reverberation is also reduced by speech conditioning circuitry and the resultant signals more reliably represent the original speech signals. Accordingly, the system provides more acceptable transmission of voice signals from noisy environments, and more reliable operation of automatic speech recognition systems. It will also be appreciated that, although a specific embodiment of the invention has been described for purposes of illustration, various modifications may be made without departing from the spirit and scope of the invention. Accordingly, the invention should not be limited except as by the appended claims.

## CLAIMS

### What is claimed is:

1. A microphone array processing system for performance enhancement in noisy environments, the system comprising:

a plurality of microphones positioned to detect speech from a single speech source and noise from multiple sources, and to generate corresponding microphone output signals, one of the microphones being designated a reference microphone and the others being designated data microphones;

a plurality of bandpass filters, one for each microphone, for eliminating from the microphone output signals a known spectral band containing noise;

a plurality of adaptive filters, one for each of the data microphones, for aligning each data microphone output signal with the output signal from the reference microphone; and

a signal summation circuit, for combining the filtered output signals from the microphones, whereby signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio.

2. A system as defined in claim 1, and further comprising speech detection circuitry, for enabling the plurality of adaptive filters only when speech is detected.

3. A system as defined in claim 1, and further comprising speech conditioning circuitry coupled to the signal summation circuit, to reduce reverberation effects in the output signal.

4. A system as defined in claim 3, wherein each of the adaptive filters includes:

means for filtering data microphone output signals by convolution with a vector of weight values;

means for comparing the filtered data microphone output signals from one of the data microphones with reference microphone output signals and deriving therefrom an error signal; and

means for adjusting the weight values convolved with the data microphone output signals to minimize the error signal.

5. A system as defined in claim 4, wherein each of the adaptive filters further includes fast Fourier transform means, to transform successive blocks of data microphone output signals to a frequency domain representation to facilitate filtering.

~~6.~~ A method for improving detection of speech signals in noisy environments, the method comprising:

positioning a plurality of microphones to detect speech from a single speech source and noise from multiple sources, one of the microphones being designated a reference microphone and the others being designated data microphones;

generating microphone output signals in the microphones;

filtering the microphone output signals in a plurality of bandpass filters, one for each microphone, to eliminate from the microphone output signals a known spectral band containing noise;

adaptively filtering the microphone output signals in a plurality of adaptive filters, one for each of the data microphones, and thereby aligning each data microphone output signal with the output signal from the reference microphone; and

combining the adaptively filtered output signals from the microphones in a signal summation circuit, whereby signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio.

7. A method as defined in claim 6, and further comprising the steps of:  
detecting speech received by the microphones; and



3 enabling the step of adaptively filtering the microphone signals only when  
4 speech is detected.

1 8. A method as defined in claim 6, and further comprising the step of  
2 conditioning the combined signals in speech conditioning circuitry coupled to the signal  
3 summation circuit, to reduce reverberation effects in the output signal.

1 9. A method as defined in claim 8, wherein the step of adaptively filtering  
2 includes:  
3 filtering data microphone output signals by convolution with a vector of  
4 weight values;  
5 comparing the filtered data microphone output signals from one of the data  
6 microphones with reference microphone output signals and deriving therefrom an error  
7 signal;  
8 adjusting the weight values convolved with the data microphone output  
9 signals to minimize the error signal; and  
10 repeating the filtering, comparing and adjusting steps to converge on a set  
11 of weight values that results in minimization of noise effects.

1 10. A method as defined in claim 9, wherein the step of adaptively filtering  
2 further includes:  
3 obtaining a block of data microphone signals;  
4 transforming the block of data to a frequency domain using a fast Fourier  
5 transform;  
6 filtering the block of data in the frequency domain using a current best  
7 estimate of weighting values;  
8 comparing the filtered block of data with corresponding data derived from  
9 the reference microphone;  
10 updating the filter weight values to minimize any difference detected in the  
11 comparing step;

12 transforming the filter weight values back to the time domain using an  
13 inverse fast Fourier transform;  
14 zeroing out portions of the filter weight values that give rise to unwanted  
15 circular convolution; and  
16 converting the filter values back to the frequency domain.

## MICROPHONE ARRAY PROCESSING SYSTEM FOR NOISY MULTIPATH ENVIRONMENTS

### ABSTRACT OF THE DISCLOSURE

5

Apparatus and a corresponding method for processing speech signals in a noisy reverberant environment, such as an automobile. An array of microphones (10) receives speech signals from a relatively fixed source (12) and noise signals from multiple sources (32) reverberated over multiple paths. One of the microphones is designated a reference microphone and the processing system includes adaptive frequency impulse response (FIR) filters (24) enabled by speech detection circuitry (21) and coupled to the other microphones to align their output signals with the reference microphone output signal. The filtered signals are then combined in a summation circuit (18). Signal components derived from the speech signal combine coherently in the summation circuit, while noise signal components combine incoherently, resulting in composite output signal with an improved signal-to-noise ratio. The composite output signal is further processed in a speech conditioning circuit (20) to reduce the effects of reverberation.

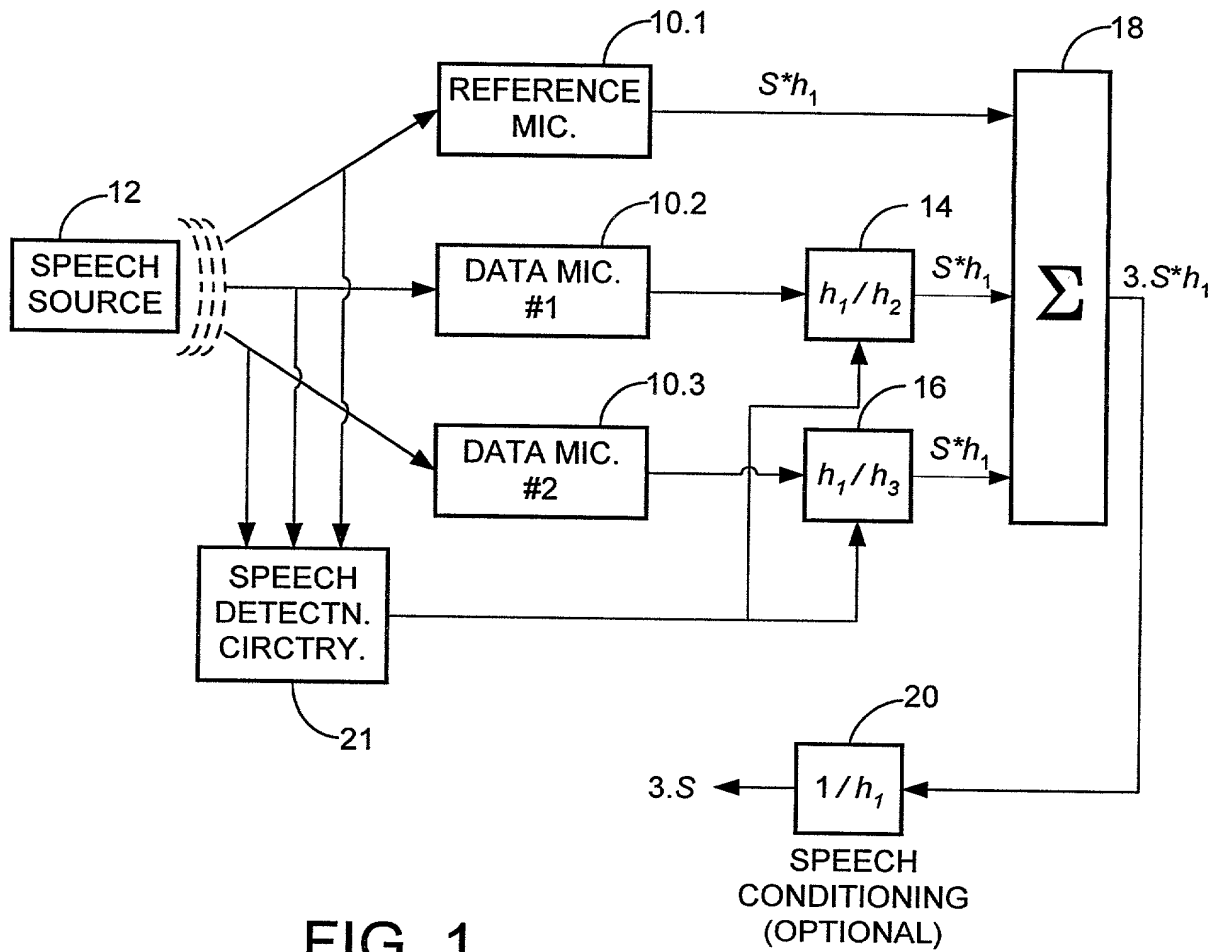


FIG. 1

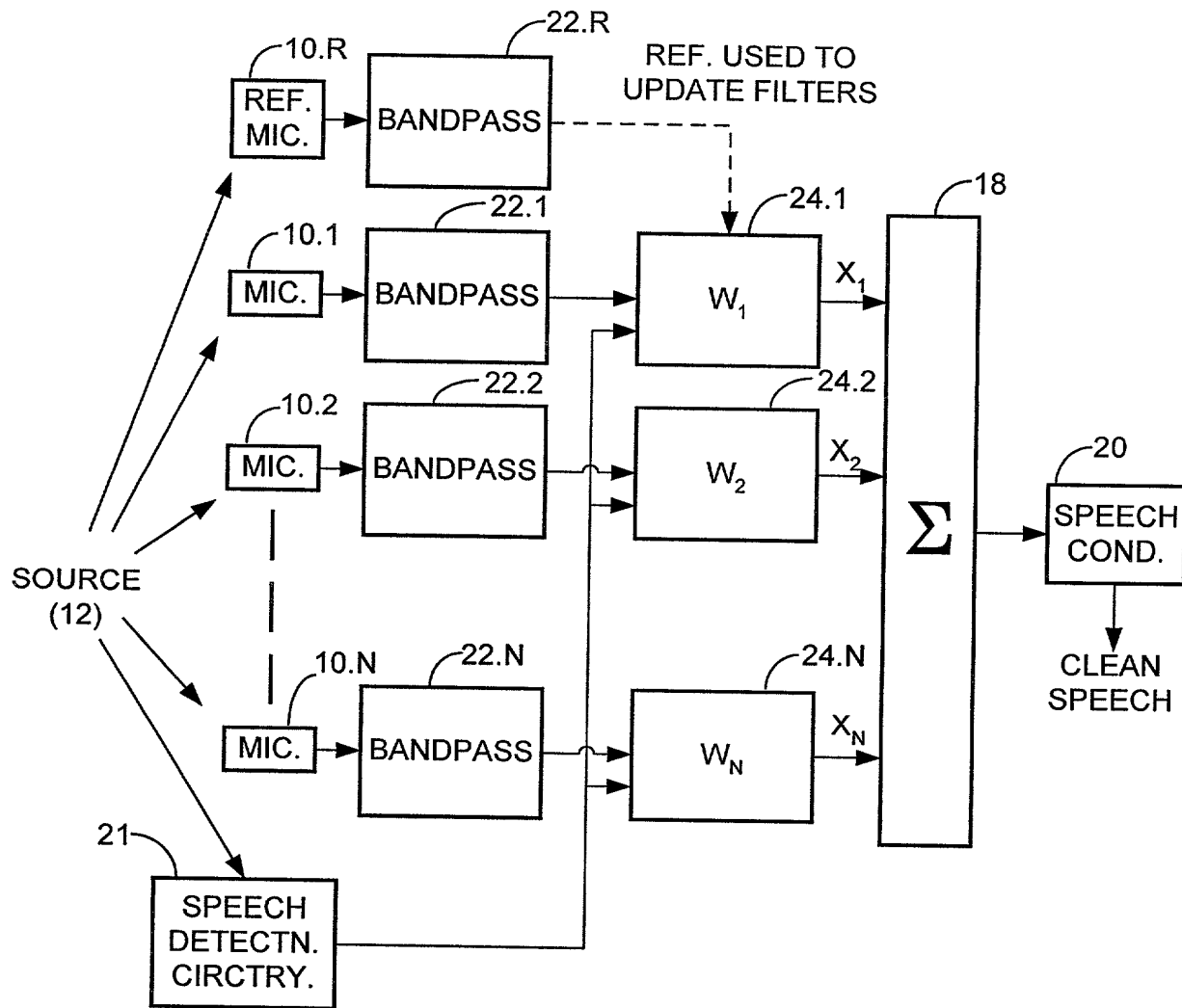
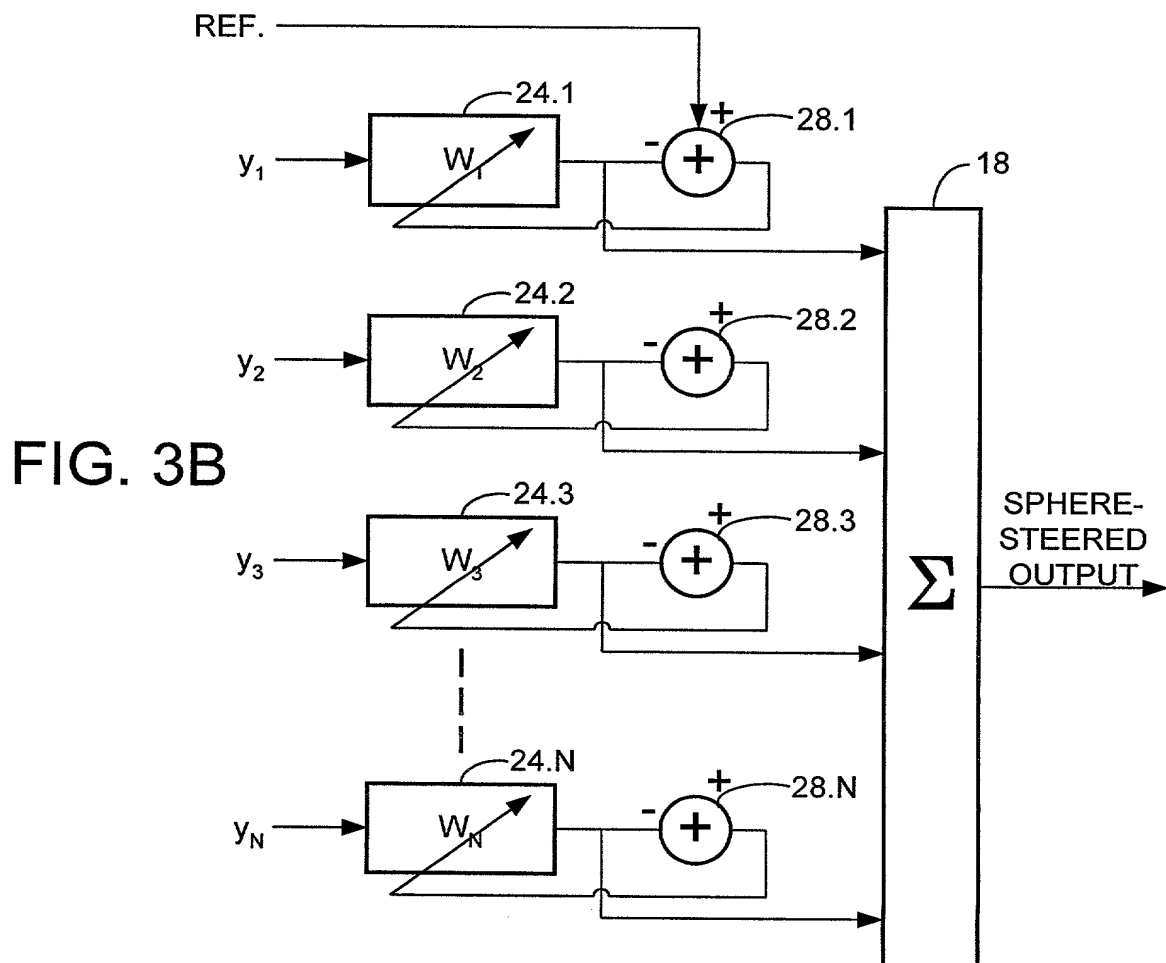
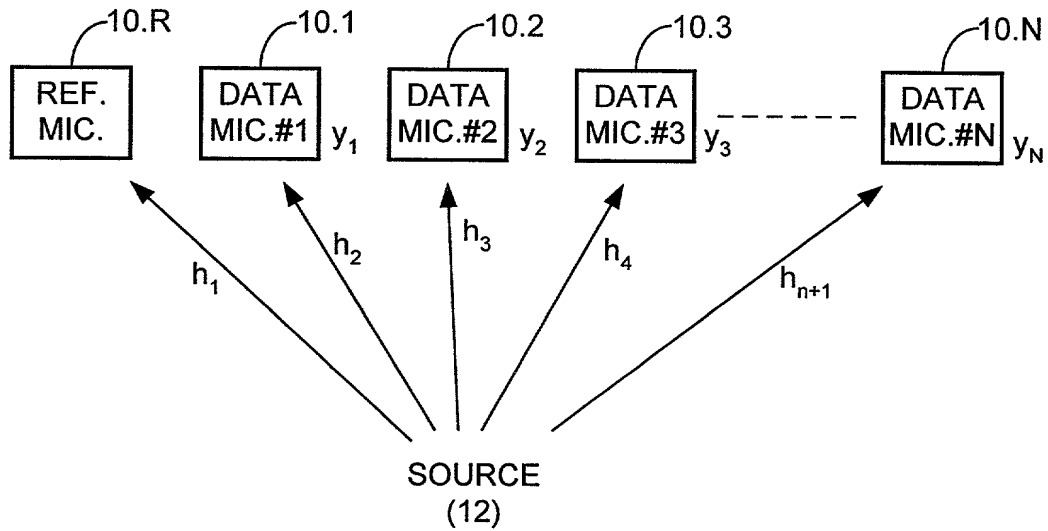


FIG. 2



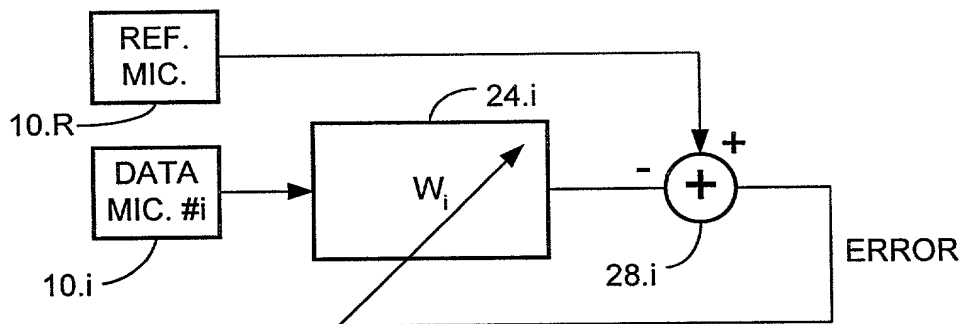


FIG. 4

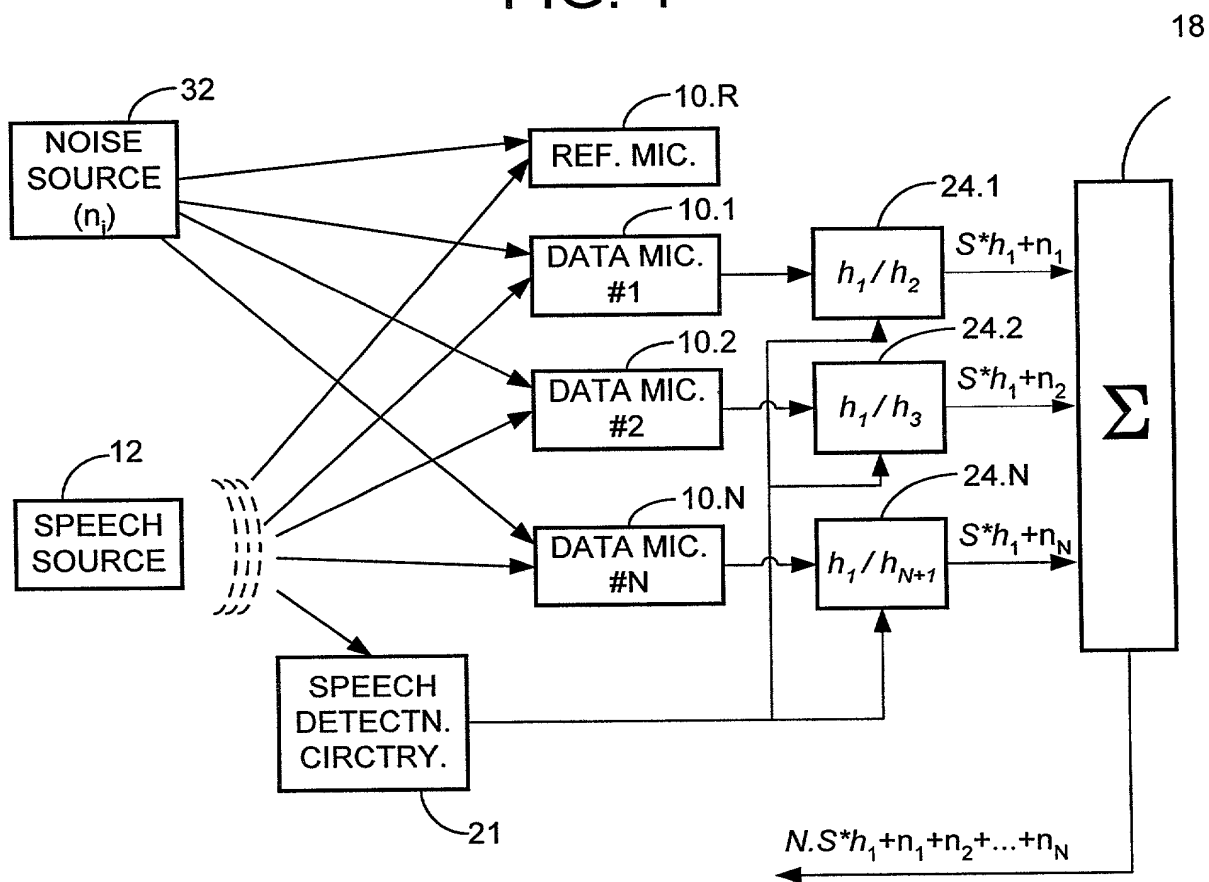


FIG. 5

Multichannel Baseline Algorithm (Linear Processing)

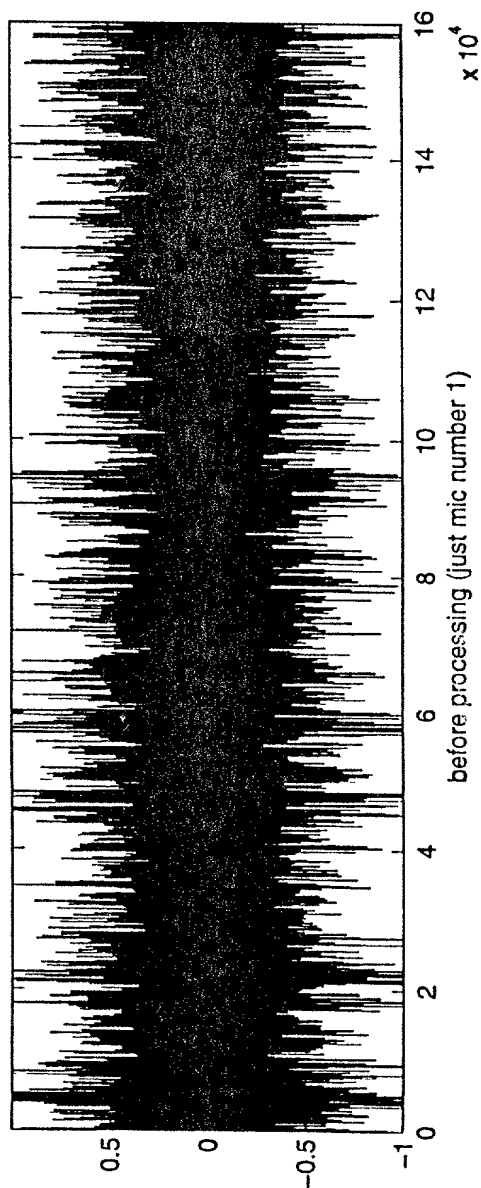


FIG. 6

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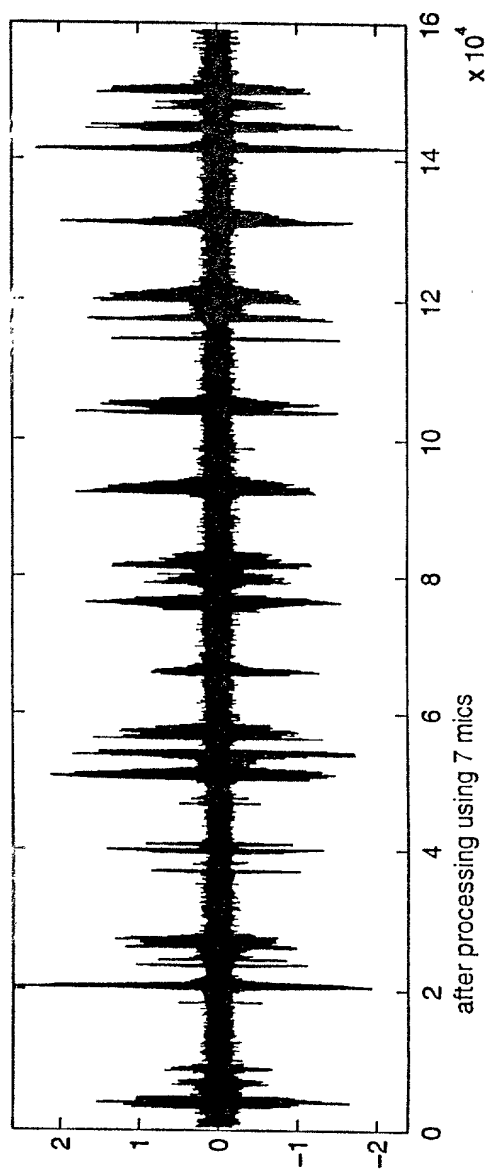


FIG. 7



Docket No. 15-0195

DECLARATION FOR PATENT APPLICATION

As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name,

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention MICROPHONE ARRAY PROCESSING SYSTEM FOR NOISY MULTIPATH ENVIRONMENTS the specification of which

X is attached hereto

\_\_\_\_\_ was filed on \_\_\_\_\_ as Application  
Serial No. \_\_\_\_\_ and was amended on  
\_\_\_\_\_.  
(if applicable)

I hereby state that I have reviewed and understand the contents of the above identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, §1.56(a).

I hereby claim foreign priority benefits under Title 35, United States Code, §119 of any foreign application(s) for patent or inventor's certificate listed below and have also identified below any foreign application for patent or inventor's certificate having a filing date before that of the application on which priority is claimed:

Prior Foreign Application(s)

Priority Claimed

NONE  
(Number)

\_\_\_\_\_  
(Country)

\_\_\_\_\_  
(Day/Mo./Yr. Filed)

\_\_\_\_\_  
Yes

\_\_\_\_\_  
No

**Docket No. 15-0195**

I hereby claim the benefit under Title 35, United States Code, §120 of any United States application(s) listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph of Title 35, United States Code, §112, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, §1.56(a) which occurred between the filing date of the prior application and the national or PCT international filing date of this application:

<u>NONE</u> (Number)	<u>                    </u> (Country)	<u>                    </u> (Day/Mo./Yr. Filed)	<u>                    </u> (Status)
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Noel F. Heal, Reg. No. 26,074

each with full power to prosecute this application, to transact all business in the United States Patent and Trademark Office connected therewith, and to appoint and revoke associate and substitute associate attorneys.

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

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